USING OF ADAPTIVE QUADRATURE EMODULATION ON ULTRASOUND TISSUE HARMONIC IMAGING

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ABSTRACT: TISSUE HARMONIC IMAGES ARE FORMED BY UTILIZING THE HARMONIC SIGNALS THAT ARE GENERATED BY TISSUE AND BY FILTERING OUT THE FUNDAMENTAL ECHO SIGNALS THAT ARE GENERATED BY THE TRANSMITTED ACOUSTIC ENERGY. TISSUE HARMONICS ARE GENERATED DURING THE TRANSMIT PHASE OF THE PULSE-ECHO CYCLE, THAT IS, WHILE THE TRANSMITTED PULSE PROPAGATES THROUGH TISSUE. WHEN THE DEPTH OF TISSUE INCREASES, THE FREQUENCY OF ULTRASOUND RECEIVED ECHOES ATTENUATES. THIS PHENOMENON IS ESPECIALLY SEVERE ON HARMONIC COMPONENTS. THE REASON IS THAT HIGH ULTRASONIC FREQUENCIES ATTENUATE MORE THAN LOW ONES. IN THE CONVENTIONAL SECOND HARMONIC IMAGING, HOWEVER, A CONSTANT ATTENUATION COEFFICIENT IS TYPICALLY ASSUMED FOR ESTIMATING THE CENTER FREQUENCY OF THE HARMONIC COMPONENTS IN QDM. THIS ASSUMPTION MAY NOT BE VALID BECAUSE THERE IS THE ATTENUATION VARIATION IN AN IMAGING AREA. TO OBTAIN THE BEST SNR, AUTOMATIC CENTER FREQUENCY ESTIMATORS BASED ON AUTOREGRESSIVE (AR) MODELING FOR THE ULTRASOUND SECOND-HARMONIC TISSUE IMAGING HAVE BEEN PROPOSED AND ARE IN USE. IN THIS PAPER WE TRY TO STUDY ABOUT USAGE OF ADAPTIVE QUADRATURE DEMODULATION FOR ULTRASOUND TISSUE HARMONIC IMAGING AND RELATED ISSUES.

KEY WORDS: ATTENUATION, DOWNSHIFT, DYNAMIC FILTER, FREQUENCY, SIGNAL-TO-NOISE

INTRODUCTION

The ultrasound echo attenuation depends on frequency, propagating depth and tissue characteristics. Thus, the attenuation dependent on frequency results in a larger attenuation of high frequencies than lower when the wave propagates through the tissue. As a result, the central frequency of the echo generates the increasing downshift with the increasing of depth. In the traditional I/Q demodulation method, it is assumed that the central frequency of the echo is the same as the transmitting frequency and unchanged all time. The assumption directly causes that the acquired I/Q signals are not perfect baseband ones but biased due to the echo attenuation. In addition, the unreasonable assumption will keep the echo from getting better signal-to-noise ratio. Quadrature demodulation method based on tracking the ultrasound echo frequency is one of the methods which is studied in this paper. The method consists of the traditional I/Q demodulator, the frequency tracking module, the phase compensation module and the dynamic filtering module. The outputs of I/Q demodulator are biased. Autocorrelation technique is utilized in the frequency tracking unit to estimate the frequency bias according to the outputs of I/Q demodulator. The estimated bias feeds to the phase compensation unit which can eliminate the frequency bias by simple trigonometric function transform. The compensated signals feed to the dynamic filter and are further processed. The bandwidth of the dynamic filter decreases with the increasing of the depth, which makes the echo acquire better SNR in different depth. According (Tranquart et al., 1999; Kim et al.,2008 ) the efficiency of the proposed method is testified by both simulations and experiments.

Here we should explain two main definitions related to quadrature demodulation. One of them is quadrature demodulation (QDM) and the other one is signal-to-noise ratio (SNR). Adaptive Quadrature Demodulation is derived from quadrature demodulation method. In this method, the estimated center frequency of a second-harmonic component is directly used for dynamic QDM and dynamic low-pass filtering (LPF). This method is capable of providing higher contrast resolution while reducing high-frequency noise under visual examination due to the improvement of SNR.
THE METHOD

In this part first we want to talk about Frequency estimation based on AR model and Adaptive quadrature demodulation based on estimation of the center frequency. An AR model is a typical example of model-based spectral estimation.

Higher order AR models may estimate the frequency more accurately at the expense of computational complexity. In Da-Young Lee et al. (2010) for the medical ultrasound applications, a 2\textsuperscript{nd} order AR model is sufficient to estimate the center frequency of echo signal.

In the following we have Fig. 1 related to the method. It shows a functional block diagram of the proposed method, i.e., the adaptive QDM based on estimating center frequency. The received echo signal \( r(n) \) is generally expressed as:

\[
r(n) = A(n) \cos(2\pi(f_0 + f_{dsh}(n))n + \varphi_1) + B(n)\cos(2\pi(2f_0 + f_{dsh}(n))n + \varphi_2)
\]

In this formulation \( A(n) \), \( \varphi_1 \), \( B(n) \), \( \varphi_2 \), \( f_0 \), \( f_{dsh} \) and \( f_{dsh} \) are the amplitude and phase of the fundamental component, those of the second harmonic component, transmitting frequency, downshift frequencies of the fundamental and the harmonic components, respectively. In the conventional QDM, \( r(n) \) is converted into in- and quadrature-phase components (i.e., \( i(n) \) and \( q(n) \), respectively) by multiplying with cosine and sine signals that have a twice of the transmitting frequency.

![Figure 1. Block diagram of the adaptive quadrature demodulation method based on estimating center frequency](image)

From the above equation, \( i(n) \) is obtained in this way:

\[
i(n) = (r(n) \cdot \cos(2\pi2f_0n)) \times h_{lpf}(n)
\]

\[
= [A(n) \cos(2\pi(f_0 + f_{dsh})n + \varphi_1) \cos(2\pi2f_0n)] \times h_{lpf}(n)
\]

\[
B(n)\cos(2\pi(2f_0 + f_{dsh})n + \varphi_2)
\]

Then we try to define \( q(n) \), so we formulate as written here.

\[
q(n) = (r(n) \cdot \sin(2\pi2f_0n)) \times h_{lpf}(n)
\]

\[
= [A(n) \cos(2\pi(f_0 + f_{dsh})n + \varphi_1) \sin(2\pi2f_0n)
+ B(n)\cos(2\pi(2f_0 + f_{dsh})n + \varphi_2) \sin(2\pi2f_0n)] \times h_{lpf}(n)
\]

\[
= -\frac{B(n)}{2} \sin(2\pi f_{dsh}n + \varphi_2)
\]

In the above equation, \( h_{lpf}(n) \) denotes the impulse response of a low-pass filter in conventional QDM.

In this method, \( i(n) \) and \( q(n) \) are divided into several subsequences with respect to time, which is used in an AR parameter estimation block. Note that the data length of the subsequence must be carefully chosen.
Since the AR model algorithm assumes that samples in the subsequences are wide-sense stationary, its data length should be small enough to satisfy the assumption. In order to eliminate the effect of data variation, a polynomial fitting is performed after estimating frequencies. The estimated center frequencies, $f_{ds}$, are used for frequency shift compensation. Also, the estimated center frequencies are used to determine the cutoff frequency of dynamic low-pass filters. As a result, the adaptive QDM provides unbiased I/Q signals, i.e., baseband signals.

**RELATED METHODS**

Ultrasound tissue harmonic signal generally provides superior image quality as compared to the linear signal but with limited penetration and the sensitivity due to low signal-to-noise ratio (SNR). The method of third harmonic ($3f_0$) transmit phasing can improve the tissue harmonic SNR by transmitting at both the fundamental and the $3f_0$ frequencies to provide mutual enhancement between the second harmonic components. To further increase the SNR without excessive transmit pressure, the phase-encoded Golay excitation can be incorporated in $3f_0$ transmit phasing to boost the tissue harmonic generation. The resultant frequency-sum and frequency-difference components of tissue harmonic signal can be simultaneously Golay-encoded for SNR improvement. Results from J. H. Chang et al. (2007) indicate that the tissue harmonic SNR increases by about 11 dB without noticeable compression artifacts.

Coded excitation also can improve the signal-to-noise ratio (SNR) in ultrasound tissue harmonic imaging (THI). However, it could suffer from the increased sidelobe artifact caused by incomplete pulse compression due to the spectral overlap between the fundamental and harmonic components of ultrasound signal after nonlinear propagation in tissues. There are three coded tissue harmonic imaging (CTHI) techniques based on band pass filtering, power modulation and pulse inversion (i.e., CTHI-BF, CTHI-PM, and CTHI-PI) which are usually evaluated by measuring the peak range side lobe level (PRSL) with varying frequency bandwidths. From simulation and in vitro studies in J. Song et al. (2010), the CTHI-PI outperforms the CTHI-BF and CTHI-PM methods in terms of the PRSL, e.g., $-43.5$ dB vs. $-24.8$ dB and $-23.0$ dB, respectively.

**CONCLUSIONS**

In this paper, we talked about adaptive QDM based on estimating the center frequency of the second harmonic component by using a 2nd order AR model. Also we talked about other related methods in this field. We found that this method is capable of estimating the change in the center frequency along imaging depth accurately. Furthermore, since this method is capable of compensating for the frequency downshift with the estimated center frequencies along depth, it could provide a better SNR.

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**REFERENCES**


